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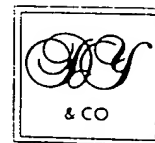
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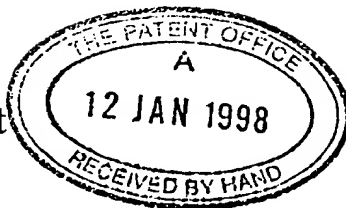
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AUDIO SIGNAL PROCESSOR

5. Name of your agent (if you have one)

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Audio Signal Processors

The present invention relates to audio signal processors for use in Cochlear Implants and to prosthesis for use as a Cochlear Implant.

Conventional hearing-aids provide considerable help for most individuals with a mild, moderate or severe hearing loss. However, these aids are of little help where the deafness is 'profound' (average loss is greater than about 100dB in both ears). In such cases an electronic device, surgically implanted in the inner-ear, can provide electrical stimulation to the nerve of hearing, giving the individual a degree of hearing sensation. In some cases open-set speech discrimination is possible, e.g. understanding a telephone conversation.

A Cochlear Implant takes-in environmental sounds, including speech, and converts this into an electrical signal which, by way of for example an implanted wire electrode array, stimulates discrete regions of the inner-ear Cochlea.

From the mid 1980s to around 1990, patients considered suitable for a Cochlear Implant were mainly adults who had, before their deafness, acquired speech and language. They were old enough to understand the implications regarding surgery and post-operative rehabilitation and, having past experience of speech and language, there was considerable potential for a return to an oral communication environment. Gradually, as clinicians around the world became more aware of the benefits of the Cochlear Implant, the focus of attention turned to the profoundly deaf child. From around 1990 onwards, an increasing number of children received a Cochlear Implant and, in the main, the results have been encouraging.

Because of the success of Cochlear Implants it is expected that, in the future, these devices will even be considered for patients having a greater amount of residual hearing.

Currently all Cochlear Implants are implemented with Digital Signal Processors. Current devices, regardless of manufacturer, are based upon digital technology, for example standard DSP chips and ASICs. The patient wears an external 'speech processor', about the size of a large match-box. This picks-up and processes environmental sounds and passes an electrical signal, via a radio-frequency link, to a 'receiving' device implanted in the ear. This internal receiver sends an electrical signal through a long thin multi-electrode array (up to 22 separate electrodes) within the inner turns of the Cochlea. Thus, the Cochlea is electrically stimulated at discrete sites and the result is a perception of sound. The stimulus intensity, delivered to each channel of the electrode array, needs to be programmed 'channel by channel'. This technology has significant advantages of flexibility, with modifications being achievable through software rather than hardware. The use of a Digital Signal Processor (DSP) provides the manufacturer with the ease of using software to alter various parameters which might be thought important in the development of new processing strategies.

A pre-requisite of all modern hearing-aids and Cochlear Implants is a method of adjustment of the intensity-frequency content of the output of the device in order to compensate appropriately, across the frequency range, for the individual's pattern of hearing loss. For any one frequency, or band of frequencies, this includes device adjustment for both the 'threshold' level of hearing and the 'uncomfortable' loudness level; the difference between these two values being known as the 'dynamic range'. In conventional hearing-aids this is accomplished by potentiometer-controlled low and high-pass analogue filtering (tone-control) in combination with 'output compression'. With Cochlear Implants, this output shaping has, up to the present time, been performed by channel-by-channel 'programming'; the reasons for this being that some intracochlear electrodes do not make good electrical contact and, even with good contact, some electrodes produce poor quality sound perception. These electrodes are 'programmed-out' and excluded from the sequence of electrical stimulation of the Cochlea.

The Cochlear Implant designs discussed hereinabove are based upon long, multi-channel electrodes, inserted deep within the Cochlea. The multi-channel design can be used to provide tonotopically distributed information from several processing strategies namely:

- i. Continuous Interleaved Sampling - CIS,
- ii. Feature Extraction or
- iii Analogue compression

Good results, in terms of open-set speech discrimination have been reported, particularly with the CIS and Feature Extraction strategies.

10

There are disadvantages associated with Cochlear Implants especially multi-channel implants:-

- i. Deep insertion of long electrodes can cause considerable damage to surviving neuronal tissue in the diseased cochlea. That is, residual hearing, albeit minimal, is destroyed.

15

- ii. The fitting/programming of current multichannel devices requires channel by-channel adjustment of stimulation levels for both threshold and uncomfortable levels. Considerable expertise is required to programme a 'MAP' which the user feels is the most useful. With current Cochlear implants, having between 12 and 22 separate electrodes, this 'channel-by-channel' programming is time-consuming, particularly since the implant has to be re-programmed about 3-4 times over the first 12 months after the operation. Some users, even with appropriate counseling, regularly attend for 'reprogramming', over several years, in the hope that one particular 'programme' will result in almost perfect hearing.

20

- iii. The DSP based technology has significant drawbacks of high power consumption and physical size. With the current digital devices batteries need changing every few (e.g. 1-2) days or even more frequently, and many patients are unhappy about wearing a relatively large speech processor. Although smaller 'behind-the ear' digital processors have reached a fairly advanced stage of development.

25

- iv. Hardware costs are high (approximately £ 15,000).

30

The use of a short electrode, single channel system has been advocated by House [7]. He argues that such a system has advantages over a 'long electrode' design in that-

- i. A short single intra-cochlea electrode will significantly reduce the possibility of damage to residual hearing.
- 5 ii. The system design is simple and relatively inexpensive (about the cost of a multichannel system)
- iii. Power consumption is low, and a head-worn processor can be used.
- iv. Fitting/programming is easier and quicker than with multichannel devices.

10

According to a first aspect of the present invention, there is provided an analogue audio signal processor for use with a Cochlear Implant and including MOS transistors operating in weak inversion.

According to another aspect there is provided an analogue tone control for use with a
15 Cochlear implant and comprising MOS transistors operating in weak inversion mode.

These aspects of our invention involve the use of analogue electronics in a way which allows realisation of an extremely small processor with a very low power requirement. Weak inversion or sub-threshold mode of operation of MOS transistors results in an
20 exponential characteristic (or a natural logarithmic characteristic) which is compatible with the exponential characteristic of the Cochlear. Although we envisage the processor being kept external (e.g. behind-the-ear), the invention does, theoretically, allow consideration of a totally implantable device. This is not true of even the most
25 modern developments in digitally-based devices. If the tone control is implanted in the Cochlear, adjustment of the frequency response is performed by wireless remote control.

According to a further aspect of the invention, there is provided a single channel audio signal processor for use in a Cochlear Prosthesis and including a tone control at least
30 the frequency response of which is controllable by the user; and /or a multi-channel channel audio signal processor for use in a Cochlear Prosthesis and including a tone

control common to all the channels at least the frequency response of which is controllable by the user.

We believe that for adults at least, and with the appropriate professional support, giving the user the ability to adjust the tonal quality of their device would be a significant step towards simplifying device re-programming after the initial fitting. We also believe that by this means the user would more readily accept the limitations of the implant and not, as is the case with some, become frustrated with the clinician's attempts at re-programming to reach a quality of sound perception which is, perhaps, for them, unachievable. To this end, our Cochlear Implant design, unlike other current designs, incorporates a 'tone-control', providing easy and rapid frequency shaping of the output. This constitutes a new innovation in Cochlear Implants. Also the use of a tone control common to all the channels of a multi-channel Cochlear Implant allows the instant and simultaneous adjustment of all the channels.

15

For a better understanding of the present invention and to show how the same may be carried into effect, reference will now be made by way of example to the accompanying drawings in which:-

20 Figure 1 is a schematic block diagram of an illustrative single channel Cochlear Implant prosthesis;

Figure 2 is a schematic block diagram of an illustrative multi-channel Cochlear Implant prosthesis;

Figure 3 is a schematic diagram of a compressor of the prosthesis of Figure 1 or 2;

25 Figure 4 is a schematic block diagram of a tone control of the prosthesis of Figure 1.1

Figure 5 is a detailed circuit diagram of an illustrative tone control ; and

Figures 6A and 6B are frequency/amplitude diagrams for the tone control of Figure 5

Referring to Figure 1 an illustrative embodiment of the invention comprises a
30 microphone 1, an amplifier and compressor 2, a tone control 3 and an implant electrode 4. The microphone produces audio signals having a dynamic voltage range

of 0 to 40 dB but the electrode 4 which stimulates the ear requires a current having a dynamic range of 2 to 15 dB. The amplifier and compressor 2 compresses the dynamic range and converts the voltage to current. The amplifier 2 may also provide volume control controllable by the user. In accordance with one aspect of the invention, the tone control 3 which is controllable by the user is provided and provides the compressed current frequency adjusted by the user to the electrode 4 which in use is implanted in the ear. This single channel embodiment of the invention operates entirely in the analogue domain. The system of Figure 1 may comprise a housing 9 containing the microphone 1, amplifier/compressor 2 and tone control 3 and which is worn by the user. As will be described with reference to Figure 5 the system can be implemented sufficiently small to allow implantation in the ear. If implanted in the ear, the tone control 3 and the volume control 2 are operated by a wireless remote control (not shown).

Referring to Figure 2 an alternative embodiment of the invention which may also operate entirely in the analogue domain is a multi-channel embodiment having an array of electrodes 7 which in use are implanted in the ear. A microphone 1, amplifier and compressor 2, and tone control 3 as described with reference to Figure 1 feed compressed and frequency adjusted audio current signals via an array of band-pass filters 5 to a CIS generator 6 which extracts Continuous Interleaved Samples which are used to stimulate the ear via the electrodes 7 in known manner.

In a further embodiment of the invention as shown in Figure 2, the single channel and multi-channel systems are combined in one apparatus, single and multi-channel operation being selectable by a switch 8. The system of Figure 2 (except for the implanted electrodes) is housed in a housing 9 worn by the user. The Cochlear Implant shown by way of example in Figure 2 is thus switchable between two system options: The single and multi-channel configurations operate in a monopolar fashion. It will be appreciated that a system may comprise only the single channel configuration. It will also be appreciated that a system may comprise only the multi channel configuration.

Referring to Figure 3, an example of the amplifier/compressor 2 is shown. The sound is picked up by the microphone 1. The compressor circuit 2 process the signal into a certain dynamic range appropriate for the specific individual. The dynamic range of the output current is controlled by a voltage controlled compressor. The dynamic range that contains most of the area of speech sounds is from about 40dB to 80dB, and, the dynamic range for electrical stimulation is narrow, in the region between 2dB to 15dB, varying from individual to individual. In order to perform the electrical compression of the signal the compressor will convert voltage to current. The

10 dynamic range of voltage (40dB) is converted into the dynamic range of current (2dB to 15dB). Here, dynamic range stands for the range between the threshold and uncomfortable level of hearing. The most appropriate circuit for this compression is the VIC (voltage-to-current converter), which allows the adjustment of the dynamic current range by means of a current control. In this case the VIC acts as a volume

15 control as well. The amplifier/compressor 2 is implemented by an MOS circuit operating in the weak inversion mode. Because the weak inversion mode is exponential (or natural logarithmic) in characteristic, it effects compression in a manner compatible with the exponential characteristic of the Cochlear.

20 The design of the illustrative Cochlear Implant prosthesis focuses on two areas :

i) Low-power electronics:

The system focuses upon a new design of analogue electronics architecture. The core of the design, that is the filters, makes use of CMOS transistors operating in weak inversion. Other branches of the circuit operate in the micro-power regime and

25 preferably in weak inversion.

ii) 'Tone-Control' for a single channel or for a multi-channel system:

In the multi-channel system the tone control is preferably common to all channels to provides instantaneous adjustment over all channels.

The tone control is based upon two low pass filters and a current subtractor.

The CMOS transistors operate in weak inversion (sub-threshold mode) and the circuit structure is based on the 'log-domain' for building the filters tunable in the audio frequency range.

- 5 Figures 4 and 5 show a micropower tone-control circuit 3 comprising two first-order log-domain filters 9 and 10 and a subtractor 11 built with MOS transistors operating in weak inversion. The tone-controller 3 is capable of providing bass cut/boost and treble cut operation as shown in Figures 6A and 6B and forms part of an all-analogue micropower Cochlear Implant.

10 The role of the tone controller is to boost/cut the low/high frequencies of the audio range. This is accomplished by the implementation of a flexible frequency shaping function which facilitates the selective placement of poles and zeros on the complex plane. Tone controllers are characterised either as passive or active [1]. Active tone
15 controllers incorporate RC-networks in a negative feedback loop around a gain block which might be an operational amplifier or a simple transistor [2]; alternatively, they may be constructed from a cascade of active high-pass and low-pass filters [3]. In this embodiment of the invention, the tone-controller is a subsystem of an all-analogue implementation of Cochlear Implant device where physical constraints such as size and
20 power consumption dictate the necessity of its implementation in an analogue micropower environment, particularly without the incorporation of conventional active (e.g. op-amps) or resistive elements. More specifically, even for a diseased Cochlea the hearing sensation depends upon the frequency of the incoming signal. For a diseased Cochlea with greater sensitivity at low frequencies than at high frequencies
25 (or vice-versa) the tone control will act to balance the hearing sensation to a comfortable level. The design of the circuit of Figures 4 and 5 is based on the recently devised log-domain design technique [4-5] which exploits the intrinsic non-linear (exponential) behaviour of a transistor and provides extended dynamic range under low power supply levels. In [6] it was shown that this technique is uniquely
30 suited for the micropower environment when MOS transistors in weak-inversion mode (or sub-threshold mode [8]) of operation are used. In addition to the wide dynamic

range possible with the log-domain technique, the design versatility offered by the implementation provides for ease and flexibility of tuning. In addition the exponential characteristic of MOS transistors operating in weak inversion and the log-domain design matches the exponential response of the Cochlea.

For the specific application the tone-controller is intended for, a bass-cut treble-cut operation is of primary importance as the controller operates in conjunction with a volume control section, i.e. the amplifier/compressor 2. Hence a "two pole - one zero" frequency shaping network is appropriate. This can be achieved by creating a pair of
 10 first-order low-pass log-domain filters 9 and 10 which are built by means of MOS transistors operating in weak-inversion and which are tuneable in the audio frequency range. The output signal is the difference 11 of the outputs of the two filters. Fig. 4 illustrates the principle and Fig. 5 shows the actual circuit comprising two first-order log-domain lowpass filters 9 and 10 and the subtractor 11.

15

Referring to Figure 5, the first filter 9 comprises a first current mirror 12 having a first current source 13 which has an input for receiving an audio signal I_{in} to be filtered and , a second current mirror 14 coupled to the first by a frequency dependent circuit in this case a shunt capacitor C_2 , the second current mirror 14 including a
 20 second current source I_{O2} controllable to vary the frequency response of the first filter.

The second filter 10 has a third current mirror 15 having a third current source 17 which has an input for receiving the audio signal to be filtered, a fourth current
 25 mirror 16 coupled to the third current mirror by a frequency dependent circuit, in this case a shunt capacitor C_1 , the fourth current mirror 16 including a fourth current source I_{O1} controllable to vary the frequency response of the second filter. The subtractor 11 forms the difference of the outputs of the first and second filters.

30 The output current $I_{out}(s)$ of the subtractor 11 is given by

$$I_{out}(t) = \frac{I_{o2} I_{b2}}{I_{d2}} - \frac{I_{o1} I_{b1}}{I_{d1}} + L^{-1} \left\{ \left(\frac{I_{o2}}{I_{d2}} - \frac{I_{o1}}{I_{d1}} \right) \frac{\left(1 + \frac{s}{\frac{I_{o2}}{I_{d2}} - \frac{I_{o1}}{I_{d1}}} \right) \left(\frac{I_{o2}(C_1 V_t)}{I_{d1} I_{d2}} - \frac{I_{o1}(C_2 V_t)}{I_{d1} I_{d2}} \right)}{\left(1 + \frac{s}{\frac{I_{d2}}{C_2 V_t}} \right) \left(1 + \frac{s}{\frac{I_{d1}}{C_1 V_t}} \right)} \right\} I_{in,a}$$

Equation 1

In Equation 1, V_t is the threshold voltage of the MOS transistors, and L is the channel width of the transistors. The meaning of the other terms is evident from Figure 5.

Eqn. (1) results in a broad passband frequency shaping network, suitable for the particular application. In the case when a tone-controller of the Baxandall type approximated by a "two-pole two-zero" function is needed, it can be implemented by feeding the input signal to the output of a log-domain lowpass 'biquad' and taking the difference as the output signal. A 'biquad' is a filter described by a biquadratic equation. The subtractor comprises transistors $M2=M3=M4=M5$ with $W = 2.4\mu m$ and $L = 2.0\mu m$, and transistor $M1$ with $W = 10\mu m$ and $L = 2.0\mu m$, for the appropriate dc output level to be realised.

The operation of the proposed circuit was simulated with SPECTRE models and AMS $2.0\mu m$ process parameters. Figs. 6A and B show the effect of the tone control at low and high frequencies. The input current is of class-A having the formula $I_{in}(t) = I_{bias}[1 + m \sin(\omega t)]$, m being the modulation index. When $I_{bias} = 10nA$ and the corner frequencies of the network is about 100 Hz and 12000 Hz, an input tone of 1000 Hz modulated by $m = 20, 30$ and 40% exhibits a THD level of -58.2 dB, -55 dB and -56.2 dB respectively. For the same corner frequencies two equal amplitude sinusoidal tones with frequencies equal to 900 Hz and 1100 Hz and modulated by $m = 40\%$ exhibited an IMD level of -46.3 dB. The total power consumption of the controller is 475nW.

Thus a specific tone controller suitable for a micropower environment has been described by way of example. The circuit comprises two log-domain lossy integrators 9 and 10 and a subtractor 11 and takes advantage of the exponential behaviour of the MOS transistors when operated in weak inversion to match the characteristics of the Cochlea.. The good dynamic range offered by the log compression coupled with flexible tuning adaptability are highly advantageous when attempting to realise an implantable analogue silicon device as a biological auditory prosthesis. The System described herein-above mainly focuses upon a new design of electronics architecture, resulting in smaller size and lower power consumption. The design is able to be applied to a multichannel CIS strategy and it also has the capability to provide a complex pulsatile stimulus to a short, single-channel electrode.

A circuit comprising MOS transistors operating in weak inversion, preferably a log domain circuit, may be used to implement the Band-pass filters 5 of Figure 2.

References

- [1] I.R.Sinclair, "Audio Electronics Reference Book", pp. 373-383 BSP
Professional Books , 1989
- 5 [2] R. F. Graf & W. Sheets, "Encyclopaedia of Electronics Circuits", Vol. 6,
pp.653, Mc-Graw Hill 1996
- 10 [3] J.Markus , "Modem Electronics Circuits Reference Manuals" , pp.6 1, McGraw
Hill 1980
- [4] D.R.Frey , "Log-domain filtering: an approach to current-mode filtering", IEE
Proceedings-G , vol. 140, pp. 406-416, 1993.
- 15 [5] D.R.Frey , "Exponential State-Space Filters: A generic current-mode design
strategy", IEEE CAS -1 , Vol. 43, No. 1, pp. 34-42, 1996
- [6] C.Toumazou, J. Ngammil and T.S. Lande, "Micropower log-domain filter for
electronic cochlea", Electronics Letters, Vol. 30, No. 22, pp. 1839-1841, 1994.
- 20 [7] House
- [8] Horowitz and Hill, The Art of Electronics 2nd Edition page 122

CLAIMS

1. A single channel audio signal processor for use in a Cochlear Prosthesis and including a tone control at least the frequency response of which is controllable by the user.
- 2 A multi-channel channel audio signal processor for use in a Cochlear Prosthesis and including a tone control common to all the channels at least the frequency response of which is controllable by the user.
- 10 3 An audio signal processor for use in a Cochlear Prosthesis and comprising a single channel processor according to claim 1, and means for generating from the output of the single channel processor a plurality of signals at respective ones of a plurality of outputs to provide a multi-channel processor and means for selecting single or multi-channel operation.
- 15 4. A processor according to claim 1 or 3, further comprising an implantable Cochlear electrode coupled to the tone control.
5. A processor according to claim 2, 3 or 4, further comprising a plurality of
20 implantable Cochlear electrodes coupled to respective ones of the said plurality of outputs.
- 6 A processor according to claim 5, wherein the generating means includes a Continuous Interleaved Sample generator.
- 25 7. A processor according to claim 3, 4, 5 or 6 wherein the selecting means comprises switching means.
8. A processor according to any preceding claim, comprising a dynamic range
30 compressor for compressing the dynamic range of the audio signal.

9. A processor according to claim 8, wherein the compressor is a voltage to current converter.
10. A processor according to claim 9, wherein the tone control receives
5 compressed current from the compressor.
11. An audio signal processor for use in a Cochlear Prosthesis and substantially as herein before described with reference to: Figure 1: or Figure 2: optionally together with Figure 3, 4 or 5 of the accompanying drawings.
- 10 12. An analogue tone control for use with a Cochlear Implant and comprising MOS transistors operating in weak inversion mode.
13. A tone control according to claim 12, comprising log-domain filtering means.
- 15 14. A tone control according to claim 12 or 13, comprising a first filter having a first current mirror having a first current source which has an input for receiving an audio signal to be filtered, and , a second current mirror coupled to the first current mirror by a frequency dependent circuit, the second current mirror including a second current
20 source controllable to vary the frequency response of the first filter.
15. A tone control according to claim 14, and further comprising a second filter having a third current mirror having a third current source which has an input for receiving an audio signal to be filtered, a fourth current mirror coupled to the third
25 current mirror by a frequency dependent circuit, the fourth current mirror including a fourth current source controllable to vary the frequency response of the second filter, and a subtractor for forming the difference of the outputs of the first and second filters.
- 30 16. A tone control substantially as herein before described with reference to Figure 5 optionally together with Figure 6 of the accompanying drawings.

17 An audio signal processor according to anyone of claims 1 to 11 and comprising a tone control as specified in claim 12, 13, 14, 15 or 16.

18 An analogue audio signal processor for use with a Cochlear Implant and including MOS transistors operating in weak inversion

19. A processor according to claim 18, comprising a voltage to current converter including MOS transistors operating in weak inversion.

10

20. A processor according to claim 19, wherein the converter is a dynamic range compressor.

15

21. A processor according to claim 19 or 20, wherein the converter is a volume control.

22. A processor according to claim 18, 19, 20 or 21, comprising a tone control including MOS transistors operating in weak inversion.

20

23. A processor according to claim 22, further comprising at least one band-pass filter including MOS transistors operating in weak inversion.

24. A processor according to claim 22 or 23 wherein the tone control and/or the or each band-pass filter is a log-domain circuit.

25

25. A filter having a first current mirror having a first current source which has an input for receiving a signal to be filtered, and , a second current mirror coupled to the first current mirror by a frequency dependent circuit, the second current mirror including a second current source controllable to vary the frequency response of the first filter.

30

ABSTRACT

AUDIO SIGNAL PROCESSORS

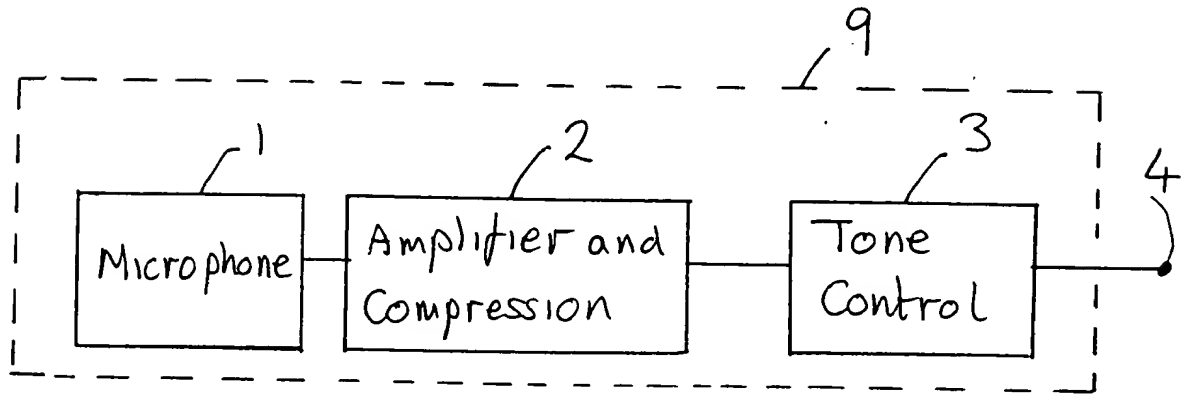
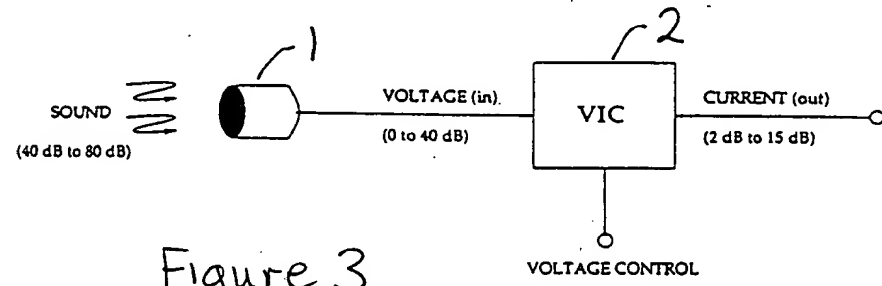
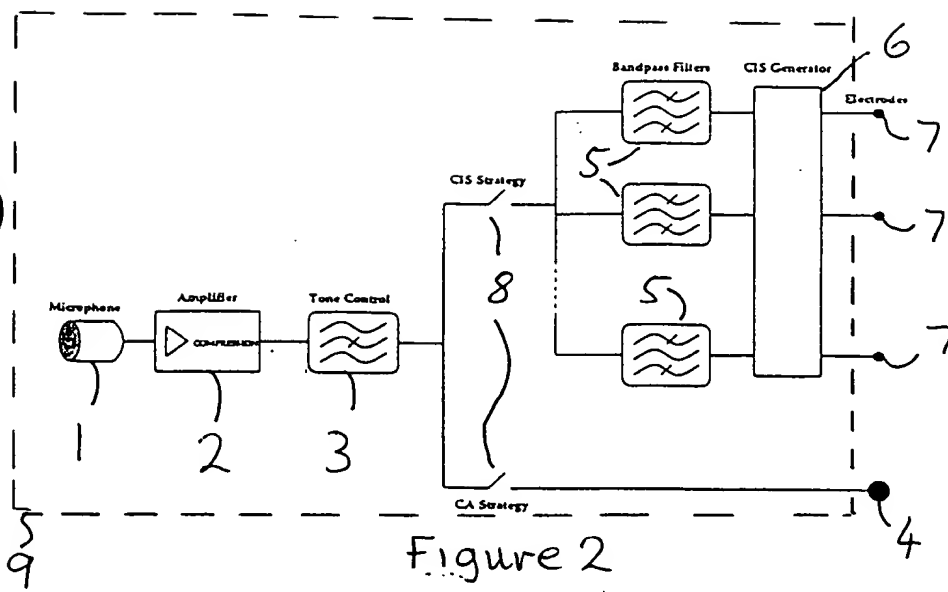
An audio signal processor for use in a Cochlear Implant comprises a single channel analogue signal processor (1,2,3) including a tone control (3) controllable by the
5 of the Implant. The tone controlled signal is fed to an implanted electrode (4).

The tone control comprises MOS transistors operating in weak inversion.

The output signal of the single channel processor (1,2,3) may be converted to multi-channel form by a converter (5,6) including for example a Continuous Interleaved Sample generator (6).

10

Figure 2



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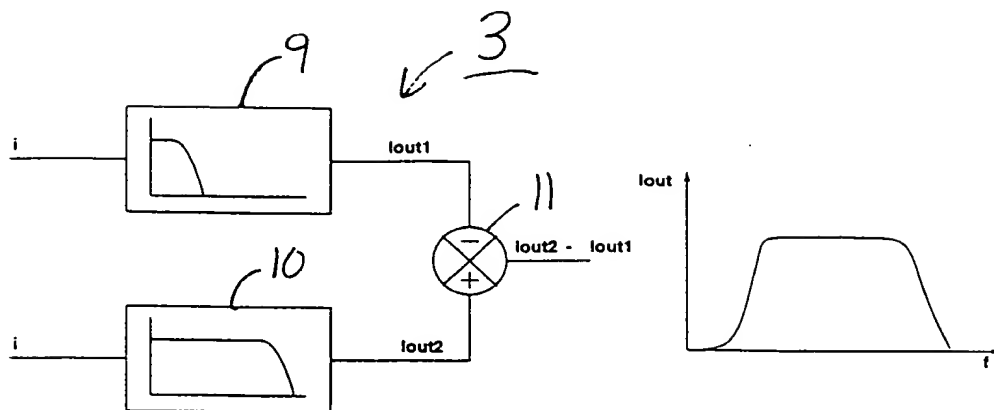


Fig 4

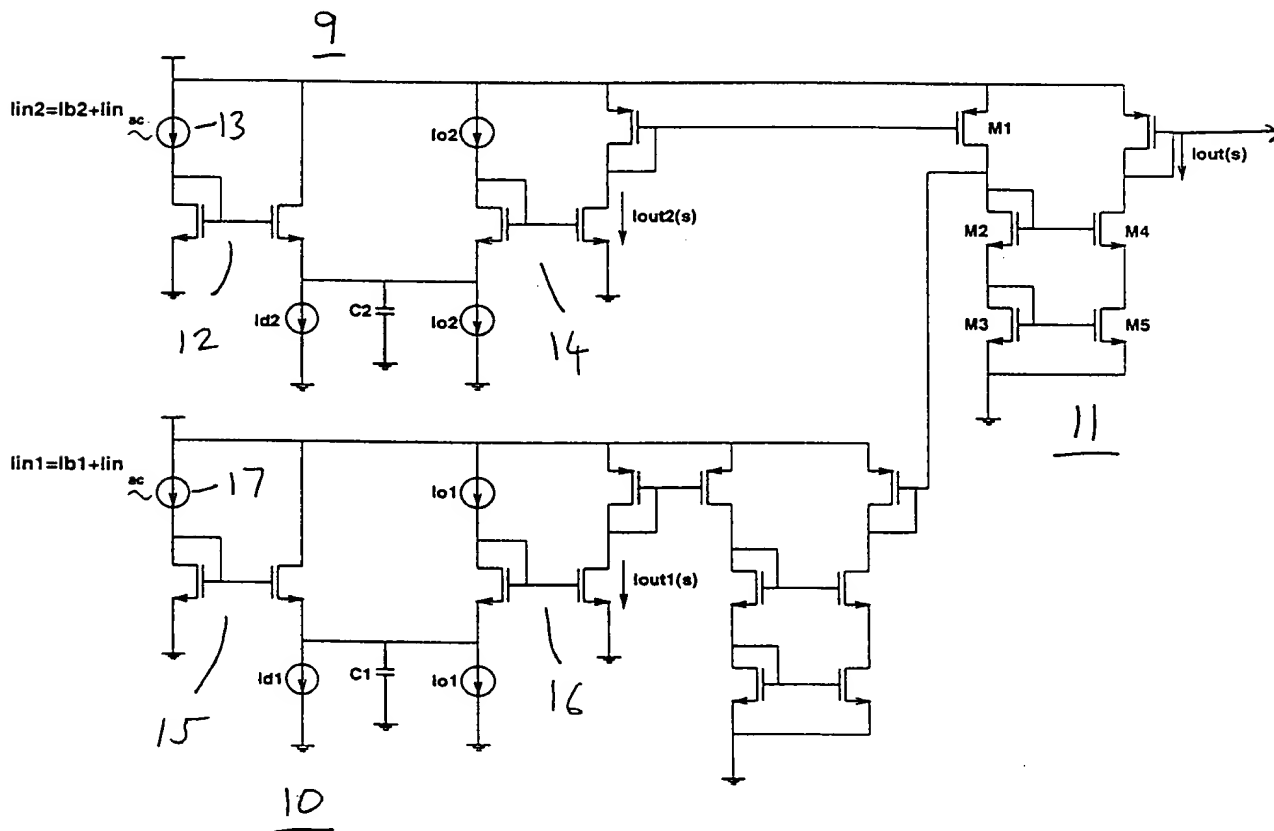


Fig 5

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Tone controller effect at bass frequencies

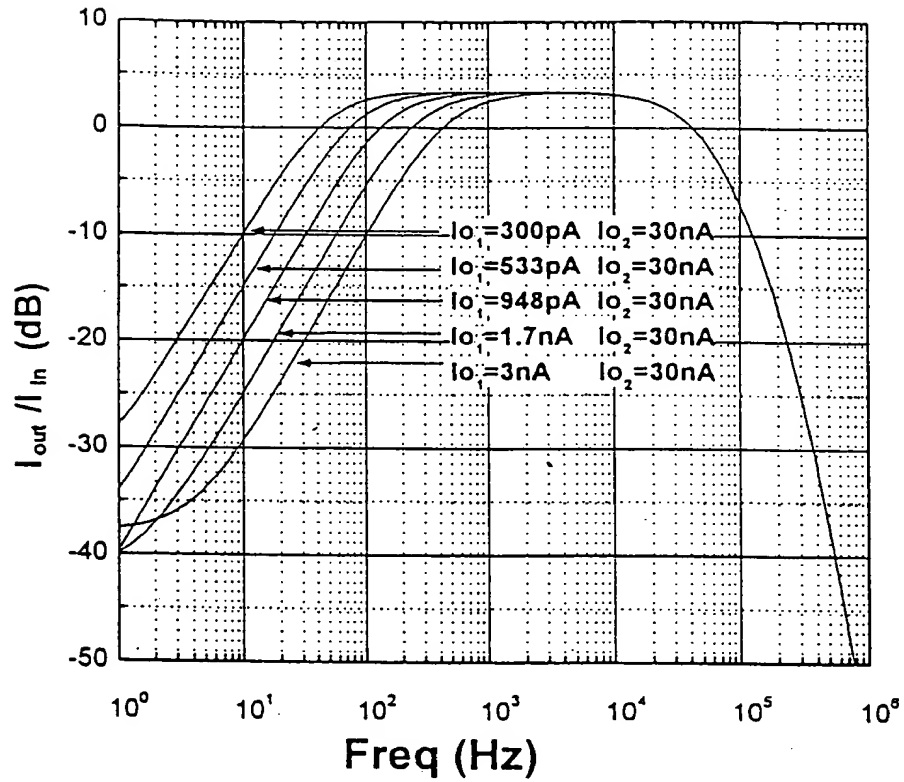


Fig 6A

Tone controller effect at treble frequencies

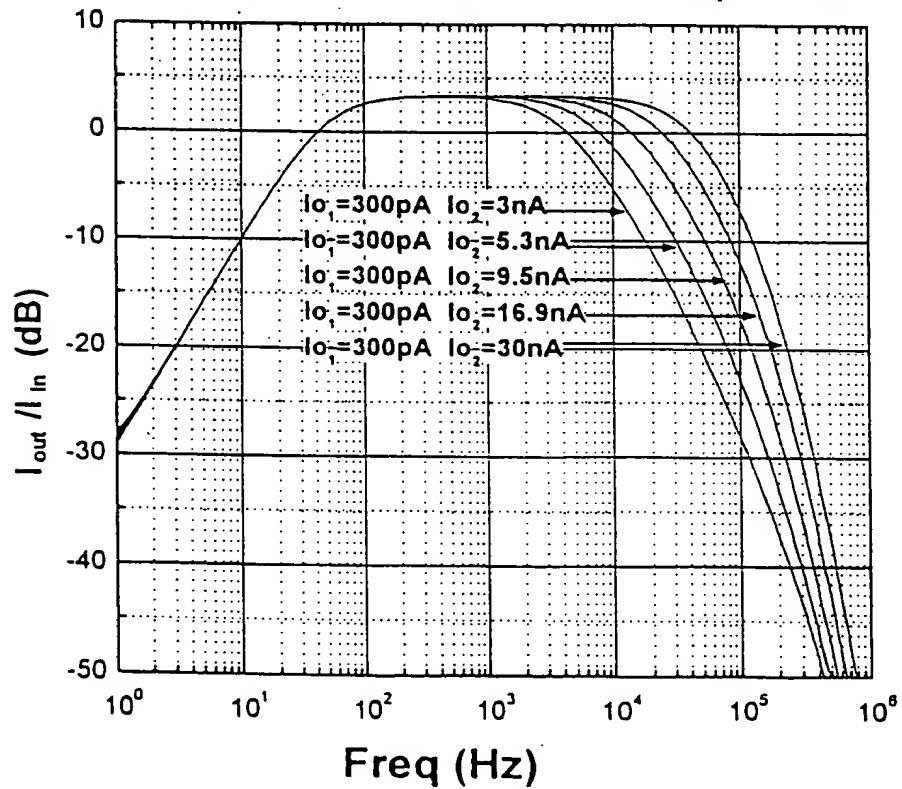


Fig 6B

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